REMARKS

The Office Action dated November 18, 2004 has been received and carefully noted. The following remarks are submitted as a full and complete response thereto.

Claim 14 has been amended and claims 15 and 16 have been added. No new matter has been added, and no new issues are raised which require further consideration and/or search. Claims 1, 4 and 6-16 are submitted for consideration.

Page 2 of the Office Action states that the Office Action is a result of the Interview Summary (paper #21) between Supervisory Primary Examiner Richemond Dorvil (for the PTO) and Applicants' representative, Arlene Neal. According to the Office Action, "the Interview Summary includes a definition of "Speech Coding" published by Arizona State University which is relied on by the applicant to narrow the term "speech coding" as it appears in the claims." Applicants respectfully but strongly submit that the Office Action simply mischaracterizes the Interview conducted between Supervisory Primary Examiner Richemond Dorvil and Applicants' representative, Arlene Neal, which resulted in an agreement that the present invention is allowable.

Specifically, the Office Action's claim that the Interview Summary "includes a definition of "Speech Coding" published by Arizona State University which is relied on by the applicant to narrow the term "speech coding" as it appears in the claim" is inaccurate. Applicants' representative submits that the Arizona State University document was not presented to narrow or define "speech coding" as suggested by the Office. Rather, the document published by Arizona State University was presented by

Applicants' representative to show that it is clear to one of ordinary skill in the art that speech coding and channel coding as defined in Kapadia et al., the prior art cited by the Office Action, are known to be two different types of coding. Channel coding is known to be different from speech coding and is not known to be another type of speech coding. Applicants further submit that during the Interview, Supervisory Primary Examiner Richemond Dorvil agreed with Applicants' represent that the documents from Arizona State University showed that speech coding and channel coding are known to be two different types of coding and that all of the pending claims are allowable. The Office Action fails to show where the Applicants relied on the document published by Arizona State University to narrow the term "speech coding" as it appears in the claim. Furthermore, there is no mention in the Interview Summary of any reliance on the document published by Arizona State University to define and/or narrow the claims of the present invention. Therefore, Applicants submit that Office Action mischaracterizes the Interview Summary of June 17, 2004 and that the Arizona State University document was not presented to in any way "narrow the term speech coding as it appears in the claims" as suggested by the Office Action.

The drawings were objected to as failing to show the respective "encoder", "decoder" and "speech coding algorithms" as relied upon by the Applicant as per the definition of Speech Coding by Arizona State University regarding the claimed "first speech coding" and "second speech coding". As noted above, the document published by Arizona State University was not presented to define and/or narrow the speech coding

claimed in the invention, but rather to point out that it is known that speech coding is different from channel coding as described in Kapadia et al. As such, Applicants submit that any interpretation of the drawings in light of a narrow definition of speech coding according to the document by Arizona State University is improper. Applicants therefore request that this objection be withdrawn.

Claims 1, 4, and 6-14 were rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,768,314 to Kapadia et al. in view of U.S. Patent Application 6,172,974 to Tseng. The rejection is traversed as being based on references that neither teach nor suggest the novel combination of features clearly recited in independent claims 1, 6, 10 and 14.

Claim 1, upon which claim 4 depends, recites a method for boosting data transmission in a telecommunications system. The method includes providing a first transmission path connecting terminal equipment with a fixed station, providing a second transmission path connecting the fixed station with a transcoder unit, transmitting speech parameters on the first transmission path using a first speech coding method, converting the speech parameters between the first speech coding method and a second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method and transmitting the speech parameters at least on a part of the second transmission path using the second speech coding method.

Claim 6, upon which claims 7-9 depend, recites an arrangement for boosting data transmission in a telecommunications system comprising a fixed station, terminal equipment, and a transcoder unit. The arrangement includes a first transmission path connecting the terminal equipment with the fixed station configured to use a first speech coding method to transmit speech parameters, a second transmission path connecting the fixed station and the transcoder unit configured to use a second speech coding method to transmit the speech parameters; at least one first speech coder configured to convert the speech parameters between the first speech coding method and the second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method.

Claim 10, upon which claims 11-13 depend, recites a mobile communications system. The system includes a base transceiver station, a mobile station, a transcoder unit and a first transmission path connecting the mobile station with the base transceiver station. The first transmission path is configured to use a first speech coding method to transmit speech parameters and at least one first speech coder is configured to convert the speech parameters between the first speech coding method and a second speech coding method. The second speech coding method is used to transmit the speech parameters on a transmission path between the first speech coder and the transcoder unit and the second speech coding method being speech coding of a lower transmission rate than the first speech coding method.

Claim 14 recites a telecommunication system having terminal equipment connected to a network side of the telecommunications system over a first transmission path configured to transmit speech parameters using a first speech coding method. The network side includes a fixed station connected to a transcoder unit over a second transmission path configured to transmit the speech parameters using a second speech coding method and a speech coder configured to receive the speech parameters from the terminal equipment and to convert the speech parameters into the speech parameters of the second speech coding method, or to receive the speech parameters to be transmitted to the terminal equipment and to convert the speech parameters into the speech parameters of the first speech coding method.

As will be discussed below, the cited prior art reference of Kapadia et al. fails to disclose or suggest the elements of any of the presently pending claims.

Kapadia et al. teaches that in a digital communications system, a speech transcoder provides the encoding and decoding ability in one component and may be referred to as a speech codec. There is also a channel codec for encoding and decoding additional information and data to the speech. Col. 1, lines 11-20. The speech codec may deliver a full rate or half rate speech to the channel codec. Col. 1, lines 21-27. The channel codec is usually at the base transceiver site, whereas the speech codec can be at the base transceiver site, a base station controller, or a mobile switching center site. Col. 1, lines 51-54. Figure 2 illustrates a communications system that may provide a combination full/half rate service type including a half rate speech codec and a full rate channel codec

arranged for communication with the half rate speech codec. Col. 3, lines 25-30. According to Figure 3, speech is delivered to an audio interface that transmits the speech to the half rate speech codec for providing coded signals. The coded signals are transmitted to the full rate channel codec for further processing. The output of the full rate channel codec is transmitted over the air to a second hybrid channel process located at a base transceiver station. A full rate channel codec of the second hybrid channel process decodes the transmitted signals and delivers them for re-ordering of the bits of the decoded signals. The re-ordered bits are then transmitted to a second half rate speech for further decoding. Col. 3, lines 35-59.

Tseng teaches wireless communication system that includes a plurality of mobile switching centers (MSC) and a plurality of base station controllers (BSC), wherein an originating MCS and BSC service an originating mobile station and a terminating MSC and BSC service a terminating mobile station. Col. 4, lines 12-31. Each MCS, BSC and mobile station provides a vocodec for performing compressing and decompressing of speech signals. Col. 4, lines 32-39. According to Tseng, the main advantage of compressing speech is that it uses less of the limited available channel bandwidth for transmission; however, there is a loss of speech quality. Col. 4, lines 55-57. This is especially true when speech is subjected to multiple instances of vocoders. Col. 4, lines 62-67. Thus Tseng provides a method and apparatus whereby compressed or modified compressed voice signals are exchanged over a transport network, such as the PSTN or ATM, and compression/decompression is only performed by the vocoders at the terminal

elements. Col. 5, lines 43-57 and Figure 1. Figure 2 illustrates a flow diagram wherein a terminating MSC or BSC generates a low frequency tone upon reception of call initiation signals from the originating MSC/BSC. The originating MSC/BSC includes circuitry for detecting the low frequency tone from the terminating MSC/BSC and for responding to the detection of this signal by generating in-band tones to disable network echo cancellers of the PSTN between the originating MSC/BSC and terminating MSC/BSC in a forward direction. The terminating MSC/BSC receives the in-band tones and generates its own in-band tone to disable network echo cancellers in the backward direction. Col. 6, lines 6-62. Figure 4 of Tseng illustrates a block diagram of a wireless communications system with improved MSC and/or BSC according to the invention.

Applicants submit that Kapadia et al. simply does not teach or suggest the features recited in claims 1, 6, 10 and 14. Claim 1, in part, recites converting the speech parameters between the first speech coding method and a second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method. Claim 6, in part, recites at least one first speech coder configured to convert the speech parameters between the first speech coding method and the second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method. Claim 10, in part recites, at least one first speech coder configured to convert the speech parameters between the first speech coding method and a second speech coding method, wherein the second speech coding method is used to transmit the speech parameters on a transmission

path between the first speech coder and the transcoder unit, the second speech coding method being speech coding of a lower transmission rate than the first speech coding method. Claim 14, in part, recites a speech coder configured to receive the speech parameters from the terminal equipment and to convert the speech parameters into the speech parameters of the second speech coding method, or to receive the speech parameters to be transmitted to the terminal equipment and to convert the speech parameters into the speech parameters of the first speech coding method.

The Office Action, on page 4, continues to suggest that Kapadia et al. teaches the step of converting between the first and second speech coding methods in figure 3. According to the Office Action, Kapadia et al. teaches that the algorithms used for the full speech rate codec and the ones proposed for the half speech codec are completely different. Hence, the parameter they produce and the parametric to sensitivity ordering are also different. As noted in the Interview Summary of the Interview on July 17, 2004, between the Examiner's supervisor and Applicants' representative, Kapadia et al. simply does not discuss or even suggest two speech coding methods wherein there exists converting means for converting the speech parameters between the first speech coding method and a second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method as recited in claims 1, 6, 10 and 14. Kapadia et al. teaches one speech coding method and a channel coding method, which are different and separate operations. In fact, Figure 3 of Kapadia et al. shows two separate operational blocks for speech coding and channel coding and

there is simply no teaching or suggestion in Kapadia et al. that these two operations could be a single operation. Moreover, in Kapadia et al. the speech coding is not changed at all. Instead, a half rate coded speech is made more robust for transmission by using full rate channel coding together with the half rate speech coding. In other words, the half rate coded speech is protected by full rate channel coding. As presented to the Supervisor Dorvil during the Interview cited in the Office Action, the speech codec (half rate codec) and channel codec (the full rate codec) of Kapadia et al. are simply not the same as the first and second coding methods of the present invention. As also presented to Supervisor Dorvil, Kapadia et al. simply does not teach or suggest two speech coding methods but rather one speech codec and a channel codec. Furthermore, it is known to those skilled in the art, that a channel codec is not the same as a speech codec as shown in the documents presented to Supervisor Dorvil, including the document from Arizona State University.

Tseng does not cure the deficiencies of Kapadia et al, in that Tseng also does not teach two speech coding methods wherein there exists converting means for converting the speech parameters between the first speech coding method and a second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method as recited in claims 1, 6, 10 and 14. The Office Action states that Tseng teaches converting compressed speech signals from one format to another intermediate common format when vocoders of the originating and terminating mobile stations are not identical. However, there is simply no teaching or

suggestion in Tseng of converting the speech parameters between the first speech coding method and a second speech coding method, the second speech coding method being speech coding at a lower transmission rate than the first speech coding method as recited in claims 1, 6, 10 and 14. Therefore, Applicants respectfully assert that the rejection under 35 U.S.C. §103(a) should be withdrawn because neither Kapadia et al nor Tseng, whether taken singly or combine, teaches or suggests each feature of claims 1, 6, 10 and 14 and hence, dependent claims 4, 7-9 and 11-13 thereon.

As noted previously, claims 1, 4 and 6-16 recite subject matter which is neither disclosed nor suggested in the prior art references cited in the Office Action. It is therefore respectfully requested that all of claims 1, 4 and 6-16 be allowed and this application passed to issue.

If for any reason the Examiner determines that the application is not now in condition for allowance, it is respectfully requested that the Examiner contact, by telephone, the applicants undersigned attorney at the indicated telephone number to arrange for an interview to expedite the disposition of this application.

In the event this paper is not being timely filed, the applicants respectfully petition for an appropriate extension of time. Any fees for such an extension together with any additional fees may be charged to Counsel's Deposit Account 50-2222.

Respectfully submitted,

Arlene P. Neal
Registration No. 43,828

Customer No. 32294
SQUIRE, SANDERS & DEMPSEY LLP
14^{TH} Floor
8000 Towers Crescent Drive
Tysons Corner, Virginia 22182-2700
Telephone: 703-720-7800
Fax: 703-720-7802

APN:mm

Enclosure: Claim Fee Sheet